SIP to SIP Basic Call Interworking

SIP-to-SIP basic functionality for IP-to-IP gateway provides termination and reorigination of both signaling and media between VoIP and video networks using SIP signaling in conformance with RFC3261. The SIP-to-SIP protocol interworking capabilities of the Cisco Multiservice IP-to-IP Gateway support the following:

- Basic voice calls (G.711 and G.729 codecs)
- Calling/called name and number
- Codec transcoding (G.711-G.729)
- DTMF relay interworking
- Interworking between SIP Fast-Start and SIP early-media signaling
- Interworking between SIP Slow-Start and SIP delayed-media signaling
- RADIUS call-accounting records
- RSVP synchronized with call signaling
- T.38 fax relay and Cisco fax relay
- TCL IVR 2.0 for SIP, including media playout and digit collection (RFC 2833 DTMF relay)
- UDP and TCP transport

Configuration Information

Configuration information is included in the Cisco Multiservice IP-to-IP Gateway Configuration Guide, Release 12.4T.

Command Reference Information

Command reference information is included in the *Cisco IOS Voice Command Reference*.

New or Modified Commands

The following commands are new or modified for this feature:

- **allow-connections**

List of All Release 12.4T Commands

An alphabetical list of all Cisco IOS Release 12.4T commands is in the *Cisco IOS Master Commands List*, Release 12.4T, at the following URL:

- http://www.cisco.com/en/US/docs/ios/mcl/124tmcl/124t_book.html

List of All New, Modified, Removed, and Replaced Release 12.4T Commands

Alphabetized lists of all new, modified, removed, and replaced commands for each Cisco IOS Release 12.4T release are in the *Cisco IOS New, Modified, Removed, and Replaced Commands*, Release 12.4T, document at the following URL:

- https://www.cisco.com/en/US/docs/ios/mcl/124tmcl/124t_nml.html